IMPROVEMENT IN METHOD FOR COMPRESSION OF SPEECH DIGITALLY CODED

Patent number:	JP1013200
Publication date:	1989-01-18
Inventor:	JIYUWAN FUUEI CHIYAN; AREN GAASHIYOO
Applicant:	BOISUKURAFUTO INC
Classification:	
- international:	G10L9/18
- european:	G10L19/06; G10L19/08G; G10L19/14P
Application number:	JP19880084973 19880405
Priority number(s):	US19870035615 19870406

Also published as:

- EP0294020 (A2)
- US4969192 (A1)
- EP0294020 (A3)

Report a data error here

Abstract of **JP1013200**

PURPOSE: To obtain a high quality speech with sufficiently low complexity for real-time coding by connecting vector quantization(VQ) to advantages of both adaptive prediction coding (ASO) and code excited linear prediction(CELP). CONSTITUTION: The VQ is connected to the advantages of both the APC and CELP. Namely, respective vectors of K speech samples are roughly calculated by using M fixed vectors stored in a VQ code book and selecting an optimum synthesized vector minimizing the size of perceptually meaningful distortion to excite a composite filter for time variation. Consequently, an analog speech or audio waveform can be coded into a bit stream which is compressed for storage and/or transmission in real time, so the waveform can be reconstitution later for reproduction and the adaptive postfiltering of a speech or audio signal which is disordered with a noise generated owing to deterioration of a coding system or other sources can be enabled so as to improve the perceptual quality of the speech or audio signal.

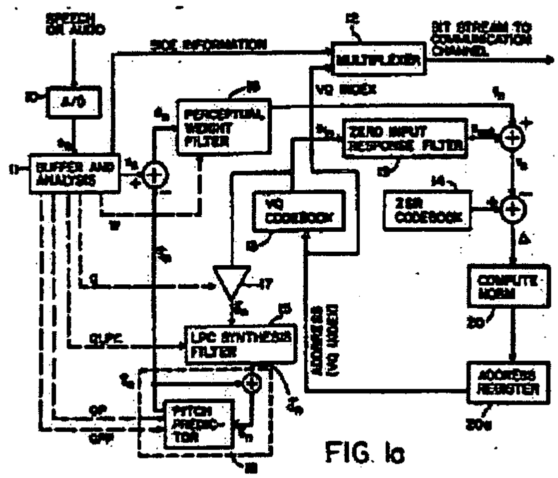

FIG. la

Data supplied from the *esp@cenet* database - Worldwide